Dolby®

Model 585

Time Scaling Processor

Issue 2   Part Number 91796

Dolby Laboratories, Inc.

Corporate Headquarters

Dolby Laboratories, Inc.
100 Potrero Avenue
San Francisco, CA 94103-4813
Telephone 415-558-0200
Fax 415-863-1373
www.dolby.com

European Headquarters

Dolby Laboratories, Inc.
Wootton Bassett
Wiltshire, SN4 8QJ, England
Telephone (44) 1793-842100
Fax (44) 1793-842101

Table of Contents

List of Figures

List of Tables

Regulatory Notices

USA

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with this instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his or her own expense.

> *WARNING:* **Troubleshooting must be performed by a trained technician. Do not attempt to service this equipment unless you are qualified to do so.**
>
> **Check that the correct fuses have been installed. To reduce the risk of fire, replace only with fuses of the same type and rating.**

Exposed portions of the power supply assembly are electrically "hot." To reduce risk of electric shock, the power cord **must** be disconnected when the power supply assembly is removed.

The ground terminal of the power plug is connected directly to the chassis of the unit. For continued protection against electric shock, a correctly wired and grounded (earthed) three-pin power outlet must be used. Do not use a ground-lifting adapter and never cut the ground pin on the three-prong plug.
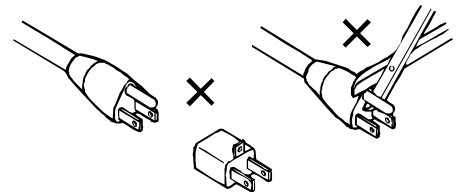
> *WARNING:* **Before applying power, check the main fuse using the procedure on page** *viii*.

Canada

This Class A digital apparatus complies with Canadian ICES-003.

EU

This equipment complies with the EMC requirements of EN55103-1 and EN55103-2 when operated in an E2 environment in accordance with this manual.

UK

Use power cord Dolby part number 92031 supplied with unit (pre-fitted with 13A plug to BS1363) to connect Model 585 to the mains supply. This equipment must be earthed.

IMPORTANT SAFETY NOTICE

This unit complies with the safety standard EN60065. The unit shall not be exposed to dripping or splashing and no objects filled with liquids, such as coffee cups, shall be placed on the equipment. To ensure safe operation and to guard against potential shock hazard or risk of fire, the following **must** be observed:

o   Ensure that your mains supply is in the correct range for the input power requirement of the unit.
o   Ensure **fuses** fitted are the **correct rating and type** as marked on the unit.
o   The unit **must be earthed** by connecting to a correctly wired and **earthed** power outlet.
o   The **power cord** supplied with this unit must be wired as follows:

<div align="right">(GB)</div>

Live—Brown   Neutral—Blue   Earth—Green/Yellow

IMPORTANT – NOTE DE SECURITE

Ce materiel est conforme à la norme EN60065. Ne pas exposer cet appareil aux éclaboussures ou aux gouttes de liquide. Ne pas poser d'objets remplis de liquide, tels que des tasses de café, sur l'appareil. Pour vous assurer d'un fonctionnement sans danger et de prévenir tout choc électrique ou tout risque d'incendie, veillez à observer les recommandations suivantes.

o   Le selecteur de tension doit être placé sur la valeur correspondante à votre alimentation réseau.
o   Les fusibles doivent correspondre à la valeur indiquée sur le materiel.
o   Le materiel doit être correctement relié à la terre.
o   Le cordon secteur livré avec le materiel doit être cablé de la manière suivante:

<div align="right">(F)</div>

Phase—Brun   Neutre—Bleu   Terre—Vert/Jaune

WICHTIGER SICHERHEITSHINWEIS

Dieses Gerät entspricht der Sicherheitsnorm EN60065. Das Gerät darf nicht mit Flüssigkeiten (Spritzwasser usw.) in Berührung kommen; stellen Sie keine Gefäße, z.B. Kaffeetassen, auf das Gerät. Für das sichere Funktionieren des Gerätes und zur Unfallverhütung (elektrischer Schlag, Feuer) sind die folgenden Regeln unbedingt einzuhalten:

o   Der Spannungswähler muß auf Ihre Netzspannung eingestellt sein.
o   Die Sicherungen müssen in Typ und Stromwert mit den Angaben auf dem Gerät übereinstimmen.
o   Die Erdung des Gerätes muß über eine geerdete Steckdose gewährleistet sein.
o   Das mitgelieferte Netzkabel muß wie folgt verdrahtet werden:

<div align="right">(D)</div>

Phase—braun   Nulleiter—blau   Erde—grün/gelb

NORME DI SICUREZZA – IMPORTANTE

Questa apparecchiatura è stata costruita in accordo alle norme di sicurezza EN60065. Il prodotto non deve essere sottoposto a schizzi, spruzzi e gocciolamenti, e nessun tipo di oggetto riempito con liquidi, come ad esempio tazze di caffè, deve essere appoggiato sul dispositivo. Per una perfetta sicurezza ed al fine di evitare eventuali rischi di scossa êlettrica o d'incendio vanno osservate le seguenti misure di sicurezza:

o   Assicurarsi che il selettore di cambio tensione sia posizionato sul valore corretto.
o   Assicurarsi che la portata ed il tipo di fusibili siano quelli prescritti dalla casa costruttrice.
o   L'apparecchiatura deve avere un collegamento di messa a terra ben eseguito; anche la connessione rete deve avere un collegamento a terra.
o   Il cavo di alimentazione a corredo dell'apparecchiatura deve essere collegato come segue:

<div align="right">(I)</div>

Filo tensione—Marrone   Neutro—Blu   Massa—Verde/Giallo

AVISO IMPORTANTE DE SEGURIDAD

Esta unidad cumple con la norma de seguridad EN60065. La unidad no debe ser expuesta a goteos o salpicaduras y no deben colocarse sobre el equipo recipientes con liquidos, como tazas de cafe. Para asegurarse un funcionamiento seguro y prevenir cualquier posible peligro de descarga o riesgo de incendio, se han de observar las siguientes precauciones:

o   Asegúrese que el selector de tensión esté ajustado a la tensión correcta para su alimentación.
o   Asegúrese que los fusibles colocados son del tipo y valor correctos, tal como se marca en la unidad.
o   La unidad debe ser puesta a tierra, conectándola a un conector de red correctamente cableado y puesto a tierra.
o   El cable de red suministado con esta unidad, debe ser cableado como sigue:

<div align="right">(E)</div>

Vivo—Marrón   Neutro—Azul   Tierra—Verde/Amarillo

VIKTIGA SÄKERHETSÅTGÄRDER

Denna enhet uppfyller säkerhetsstandard EN60065. Enheten får ej utsättas för yttre åverkan samt föremål innehållande vätska, såsom kaffemuggar, får ej placeras på utrustningen." För att garantera säkerheten och gardera mot eventuell elchock eller brandrisk, måste följande observeras:

o   Kontrollera att spänningsväljaren är inställd på korrekt nätspänning.
o   Konrollera att säkringarna är av rätt typ och för rätt strömstyrka så som anvisningarna på enheten föreskriver.
o   Enheten måste vara jordad genom anslutning till ett korrekt kopplat och jordat el-uttag.
o   El-sladden som medföljer denna enhet måste kopplas enligt följande:

<div align="right">(S)</div>

Fas—Brun   Neutral—Blå   Jord—Grön/Gul

BELANGRIJK VEILIGHEIDS-VOORSCHRIFT

Deze unit voldoet aan de EN60065 veiligheids-standaards. Dit apparaat mag niet worden blootgesteld aan vocht. Vanwege het risico dat er druppels in het apparaat vallen, dient u er geen vloeistoffen in bekers op te plaatsen. Voor een veilig gebruik en om het gevaar van electrische schokken en het risico van brand te vermijden, dienen de volgende regels in acht te worden genomen:

o   Controleer of de spanningscaroussel op het juiste Voltage staat.
o   Gebruik alleen zekeringen van de aangegeven typen en waarden.
o   Aansluiting van de unit alleen aan een geaarde wandcontactdoos.
o   De netkabel die met de unit wordt geleverd, moet als volgt worden aangesloten:

<div align="right">(NL)</div>

Fase—Bruin   Nul—Blauw   Aarde—Groen/Geel

Fusing Information

> *WARNING:* **To reduce the risk of fire, replace fuses only with the same type and rating.**

The unit uses a universal switching power supply that handles the full range of nominal mains voltages between 90 and 264 VAC and any frequency between 50 and 60 Hz.

Check Main Fuse

The Main fuse rating is:

> T 1A L (1 Amp, 250 V, 20 mm, time-lag, low-breaking capacity) for all operating voltages.

> *WARNING:* **The power to the unit must be off when the following steps are performed. Ensure that the main power cable to the unit is not connected to a power source.**

1. Open the fuse compartment door in the AC power input housing with a small flat-blade screwdriver (Figure 1). Carefully pull out the fuse carrier.

2. Check that the fuse has the correct rating. The fuse carrier must be inserted into the compartment with the orientation shown in Figure 1. *Do not force the carrier into the compartment or both could be damaged.*

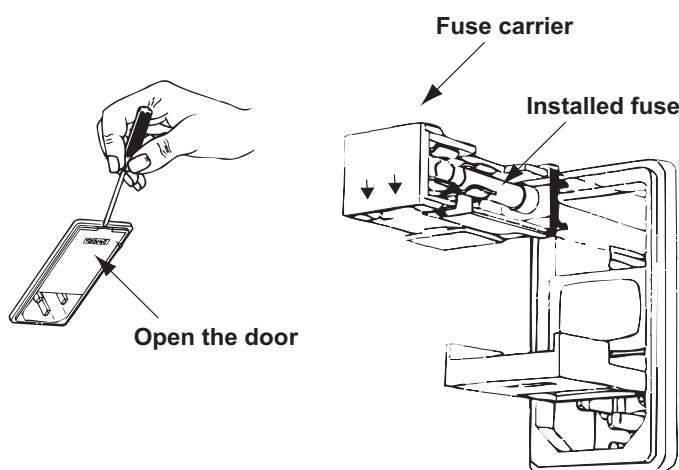3. Snap the fuse compartment door closed.

Figure 1 Checking the Main Fuse

Internal Fuse

The switching power supply contains a separate fuse. Most fault conditions should be protected by the main fuse.

If you find it necessary to replace the internal fuse, be certain to replace it with a fuse of the same type and rating as printed on the switching power supply board.

Chapter 1
Introduction

The Model 585 Time Scaling Processor delivers true time scaling and pitch shifting for the studio professional.

A change in the playback speed of analog tape—or in the sample playback rate for digital formats—has always brought about a change in the audio pitch as well as the duration of the audio program being played. However, this fixed link between playback speed and pitch has often meant a compromise in audio quality. Therefore, the ability to precisely determine and alter pitch and program duration **independently** has been a necessity since it first became possible to alter the speed of a recording.

Today, many standard operations result in a change in audio pitch; for example, when converting a film for broadcast at 25 fps, or adjusting a program's length to match the available broadcast time slot.

There have been attempts over the years to provide pitch shifting and time scaling. Early products showed glaring weaknesses, and more recent attempts have tended to be useful only on a limited range of material. Model 585 delivers dependable processing of any signal, across the whole spectrum of audio: music, speech, special effects, and any combination of these elements.

Multichannel Delivery

Model 585 is the first product to deliver truly phase-synchronous time scaling and pitch shifting for multiple (up to eight) channels of audio—something multiple stereo processors cannot achieve. It is important to preserve the phase relationship of multichannel audio programs for all listeners; it is especially important for those with stereo or mono reproduction equipment, who hear a summing of the multichannel elements. Model 585 is compatible with all Dolby® E products.

Real-Time Time Scaling and Pitch Shifting

Model 585 delivers two separate functions: time scaling and pitch shifting. Each function can be applied in real time.

Time scaling alters the duration of an audio signal but leaves the perceived pitch unchanged from the original signal. Time scaling and how to implement it using Model 585 is discussed in Chapter 3.

Pitch shifting, sometimes referred to in other documents as pitch scaling, alters the perceived frequency of the audio signal from that of the original signal, while leaving the duration of the signal the same as the original. Pitch shifting and its implementation using Model 585 is covered in Chapter 4.

Chapter 2
Getting Started

This chapter covers connection requirements for Model 585 and the basics of using the front-panel controls.

2.1   Unpacking and Inspection

Before unpacking Model 585, inspect the outer carton for shipping damage. If the carton shows damage, inspect Model 585 in the associated areas. The BNC connectors on the rear panel are covered with protective plastic caps.

The following items are provided with Model 585:

- Three power cords: one each for US, continental Europe, and the UK. Use the cord appropriate to your location.
- A bag containing rackmount screws and washers, plus five 75Ω BNC terminators.
- Warranty information. Fill out the warranty card and return to Dolby Laboratories. To receive software upgrades, register your product on the software upgrades page at www.dolby.com/download/softreg.

2.2   Installing

Installing Model 585 requires:

- Two standard rackspace units.
- Digital I/O. The digital input and output connections on Model 585 may be unfamiliar to a first-time user of a digital audio device. The Audio Engineering Society (AES) created a standard electrical interface for digital audio with a balanced XLR connector and 110Ω impedance called AES3-1995. This interface was later expanded to include an unbalanced BNC connector, 1 Vp–p signal level, and 75Ω impedance; it is now known as both AES3-ID-1995 and SMPTE-276M. Model 585 uses this new standard for digital audio electrical interface connections.

Note: These signals must be properly terminated with a 75Ω impedance at one point only. We recommend a standard video terminator.

- Cables. We recommend standard 75Ω BNC cables for digital audio signal connections. To connect to digital equipment with 110Ω XLR connectors, use impedance-matching transformers (readily available from Canare, Neutrik, and other manufacturers).

- Input channel 1/2 must be connected to a valid digital audio signal for proper operation of the unit.

Model 585 can be used between any two products that carry PCM audio. For specific installation examples, see Chapter 5, Applications.

2.3   Rear Panel

The main fuse rating is:

T 1A L (1 amp, 250 V, 20 mm, time-lag, low-breaking capacity) for all operating voltages.

⚠ *WARNING:* **Before applying power, check the main fuse, using the procedure on page** *viii***.**

There is no power switch on Model 585. To apply power, connect the power cord to a live outlet.

2.3.1   Digital Audio Connections

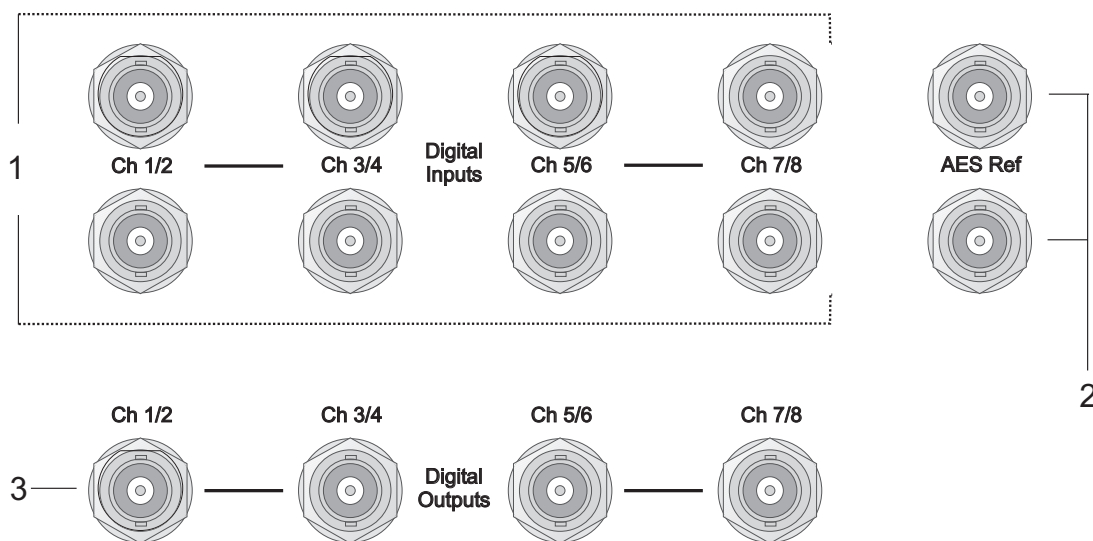Figure 2-1 shows the digital audio input and output connections.

Figure 2-1 Connections: Signal In and Out

Figure 2-1 definitions:

1. **Digital Inputs.** Model 585 requires PCM audio input at a sample rate of 48 kHz ± 15%. Each input has two connectors; either connector can be used for the input or as a loop-through for that channel pair. If the loop-through connection is not being used, it must be fitted with a 75Ω termination. The last device in the loop-through signal chain must be terminated with 75Ω.

Note: **Ch 1/2** requires a valid PCM digital audio signal. That input is used to derive the clocks for all input channels and is the reference clock for scaling.

---

2. **AES Ref.** Connect to an AES digital audio reference source. The second connector can either be used as a loop-through or be terminated, as appropriate.

Note: **AES Ref** requires a valid AES digital audio signal. That input is used to derive the clocks for all output channels and so is the reference clock for the digital outputs.

---

3. **Digital Outputs.** These connections carry the processed PCM signal.

2.3.2 Serial Connection Ports

Figure 2-2 shows the serial connection ports.

1 — Remote RS-232   Metadata In — 2

3 — GP I/O   Metadata Out — 4
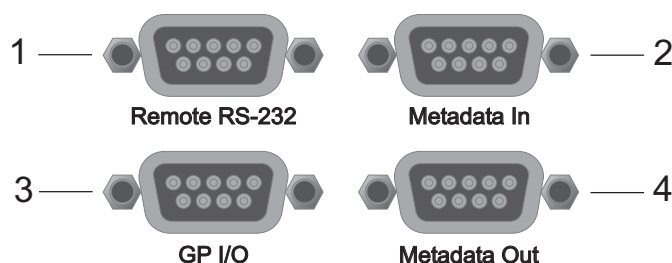
Figure 2-2 Connections: Remote RS-232, GP I/O, Metadata In and Out

Figure 2-2 definitions:

1. **Remote RS-232.** Connect to this port to update Model 585 software.

2. **Metadata In.** (Reserved for future use.) Input port for external metadata. Connect to a device that delivers Dolby® metadata, such as a Dolby E decoder or Model DP570 Multichannel Audio Tool.

3. **GP I/O.** This port provides a fixed, TTL level-sensitive output indicating the current status of Model 585.

4. **Metadata Out.** (Reserved for future use.) Output port for metadata. Connect to a downstream Dolby E or Dolby Digital encoder.

2.3.3   Remote Connection Ports

The **Remote (9-pin) In** and **Remote (9-pin) Out** ports are reserved for future enhancements.

2.4   Front Panel

The front panel is shown with feature definitions in Figure 2-3.
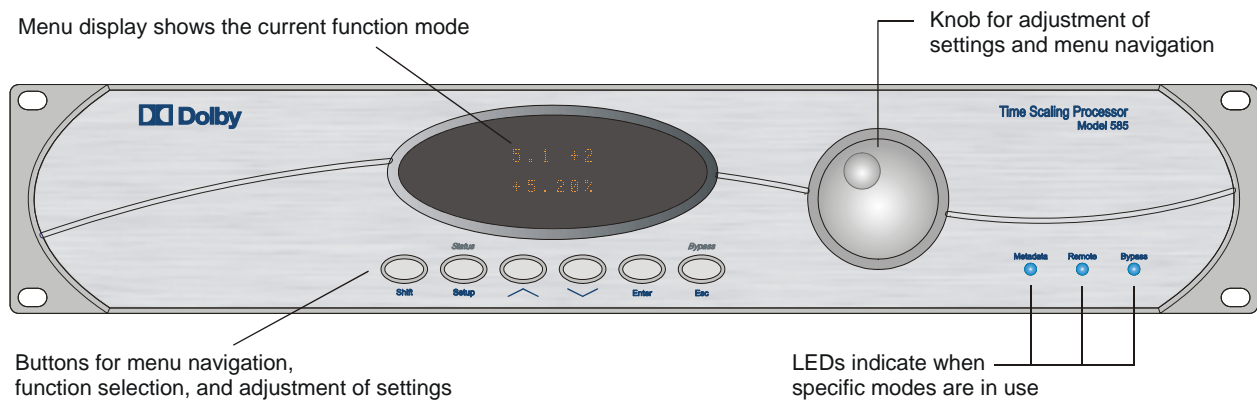
Figure 2-3 Model 585 Front Panel

The menu system has two independent structures: setup and status. The setup menu lets you control and configure Model 585 to suit your requirements. The status menu displays information regarding the operating status.

There are two ways to adjust settings on Model 585. All functions are adjustable using the buttons below the display window, and some functions can be easily adjusted using the knob.

2.4.1   Reading the Display

After powering up, Model 585 displays the main status menu.

The factory default is:

```
      5.1 + 2              Sens=5
                   +0.0
```

The main status menu shows you:

- On the upper line, the program configuration and the sensitivity level. In this example, `5.1 + 2` defines two programs: Program 1, a 5.1 mix; and Program 2, a two-channel mix. The sensitivity level is set to 5.

- The amount of pitch shifting on the lower line, `+0.0%`, indicates no shift.

2.4.2   Buttons

The six buttons in Figure 2-4 display window control navigation through the menus and selection of parameters within menu screens. Press any button to use the command listed below it.

Figure 2-4 Front-Panel Buttons

Note: In the `Transport` menu, some buttons are redefined to trigger functions not available in all other menus. For more details, see Section 3.1, Transport Menu.

Table 2-1 Front-Panel Button Definitions

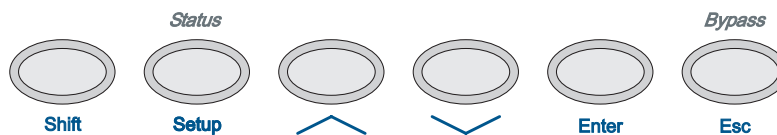| Button	Result
Shift	Selects the alternate function for a button. To select *Status* or *Bypass*, hold down **Shift** and press the button under the selection you want.
Setup	From any status menu, displays the last setup menu viewed, or the `Unit Setup` menu if a setup menu has not been viewed in the previous five minutes. From any setup menu, displays the `Unit Setup` menu.
Status	From any setup menu, displays the last status menu viewed, or the main status menu if a status menu has not been viewed in the previous five minutes. From any status menu, displays the main status menu (program configuration and amount of pitch shift).
⌃	Scrolls to previous item in menu selection display. In menus that determine a numeric value, increases that value.
⌄	Scrolls to next item in menu selection display. In menus that determine a numeric value, decreases that value.
Enter	Activates the menu or setting on the lower line of the display.
Esc	Reverts to the next-higher menu level, or if the active setting is different from the setting on screen, returns the display to the active setting.
Bypass	Activates bypass mode. Processing latency remains in effect, but no adjustments are made to the output signal. To take the unit out of bypass mode, press **Shift** + *Bypass* again.

Note: In all menus, settings other than the active one display a flashing block next to the setting. Press **Enter** to select the setting shown, or **Esc** to return to the active setting.

2.4.3   Knob

The knob offers quick adjustment of many menu options.

Rotating clockwise scrolls through setup menu options in the same way as pressing ⌃. Rotating counterclockwise scrolls through setup menu options in the same way as pressing ⌄.

In the main status menu, you can adjust the pitch shift value by turning the knob. Pressing **Shift** while turning the knob changes the sensitivity level. Pressing either arrow button scrolls to other status menu displays.

Caution :   *Adjusting the pitch shift value or sensitivity level in the main status menu*
activates that value change immediately; pressing **Enter** *is not required.*
You do not have the option to press **Esc** *to return to the original setting, so*
note what that setting is before rotating the knob.

In the `Play` menu, the knob triggers different functions than in all other menus. See
Knob Function in Play Menu on page 3-4 for details.

2.5   Status Menu

The main status menu, showing the current program configuration and pitch shift
percentage, appears when you turn the power on. At power-up, the status returns to
the configuration that was active at the preceding power-down.

When you press the *Status* button combination (**Shift** + *Status*) while in a setup menu,
the display returns to the last status menu displayed within the previous five minutes.
If that is not the main status menu, you can view that menu by pressing **Esc,** or the
Status button combination again.

In the main status menu, you can view more information about the current settings by
using the arrow buttons. Press ⌄ to view the next choice, or ⌃ to view the
previous choice. Table 2-2 defines all the status menus.

Table 2-2 Status Menu Screens

Menu	Definition
Main Status	Shows program configuration and pitch shifting value.
Firmware Version	Shows version number of firmware loaded on Model 585.
Digital In	Enter to see status of the input signal for each channel pair.
Reference In	Shows status of the AES Reference signal.

Note: There is no indication of the condition of the reference or digital input signals
on the front panel or the main status menu. The status of these signals is only
available through the individual status menus.

2.6   Setup Menu

Options in the setup menu are:

- `Program Config`
- `Operating Mode`
- `Transport`
- `System Settings`

Press **Enter** to view the menu for the option on the second line of the display. The `Transport` menu operates the time scaling function, and is described in Section 3.1, Transport Menu. The other three menus are described in this section.

2.6.1   Program Configuration

The `Program Config` menu initially displays the active program configuration. Rotate the knob or press an arrow button to scroll through the other configuration options. To process the input signal properly, the `Program Config` setting must match the configuration of the input signal.

The `Program Config` setting describes how the eight input channels are configured. The notation conventions are as follows:

The + symbol separates individual audio programs within a configuration. For example, `5.1 + 2` represents one 5.1-channel and one stereo program—which may or may not be related. The × symbol shows the number of a given channel format within that configuration. For example, `4 × 2` represents four two-channel programs—again, which may or may not be related. One configuration has four different programs: `4 + 2 + 2 × 1,` which is one four-channel, one two-channel, and two mono programs.

The `Program Config` parameter instructs Model 585 which input channels to process together as a single, phase-coherent program. For example:

`5.1`   Processes the first six input channels as one complete 5.1-channel audio program.

`5.1 + 2`   Processes the first six input channels as one complete 5.1-channel audio program. Processes channels 7 and 8 as one complete stereo program.

`4 × 2`   Processes channel pairs 1/2, 3/4, 5/6, and 7/8 as four separate stereo programs.

Model 585 processes all program configurations to the same high quality. Therefore, there is no difference in audio performance between `2 + 2`, `3 × 2`, or `4 × 2` program configurations.

Channel assignments within each program configuration follow standard mapping, as shown in Table 7-1 in Chapter 7, Reference Data. The only exception is the low-frequency effects (LFE) channel, which can also be assigned manually, as described in Section 2.6.2, Operating Mode.

2.6.2   Operating Mode

In the `Operating Mode` menu, you have six options: `Sensitivity`, `Pitch`, `Processing Delay`, `Clock Source`, `DC Filter`, and `LFE Channel`.

Sensitivity

Model 585 is able to perform high-quality time scaling of complex audio material because, in a manner comparable to that used by human listeners, it analyzes and identifies the individual audio components and processes them independently. The `Sensitivity` setting allows you to adjust the level of processing detail Model 585 applies to the audio material. The default setting, `5-high`, provides the most detailed processing and is appropriate for nearly all material.

For some material, however, the `5-high` setting may be too sensitive, causing Model 585 to focus on subtle audio elements that can interfere with the processing of the most prominent material. In this case, Model 585 may not produce its best results. For example, if material with a live audience is being processed and the `Sensitivity` setting is too high, Model 585 may focus on elements of audience or ambient sounds to the detriment of the primary audio material.

As you decrease the `Sensitivity` setting, less prominent elements of the audio material are processed less intensively, reducing the impact on the most prominent elements. If the `Sensitivity` setting is too low, however, important elements of the audio may not be processed correctly.

The `Sensitivity` setting can be adjusted in real time using the front-panel controls.

If you alter the setting using the `Sensitivity` submenu, you are able to "cue up" different settings. For example, you can be processing audio at one setting, select a new setting and then press enter to activate the new setting as required.

In addition to adjusting this setting directly within the `Sensitivity` submenu, you can also make the adjustment from the main status menu by rotating the knob while pressing **Shift.** If you do so, however, the adjustment is immediately activated, and you cannot return to the previous value by pressing **Esc.**

Note: `Sensitivity` should remain at `5-high` for most program material. If the setting must be changed, the change should be limited to the length of the specific passage that requires the change, and then returned to `5-high.` Setting changes activate in real time when changed using the front-panel controls.

Pitch

You can use the `Pitch` menu to select a new scaling value by pressing the arrow buttons or rotating the knob. You can also make this adjustment from the main status menu by rotating the knob. If you do so, however, the adjustment is immediately activated, and you cannot return to the previous value by pressing **Esc.**

Processing Delay

The `Processing Delay` menu defines the system latency in milliseconds, within a range of 400–480 ms. The default is 440 ms.

Clock Source

In the `Clock Source` menu, you can choose the source for the digital audio output clock. The default is `Digital Input,` which locks the output clock to the channel 1/2 input. `Internal 48 kHz` locks the output clock to an internal 48 kHz oscillator. `External Ref-In` locks the output clock to the **AES Ref** input clock. The status of the **AES Ref** input clock is displayed in the `Reference In` status menu.

DC Filter

The `DC Filter` can be turned on to eliminate any inconsistency due to the processing of very low frequency or subsonic sounds. The default is `Disabled.`

LFE Channel

In the `LFE Channel` menu, you can adjust the assignment of the LFE channel to match the input. The factory default is `Auto`, which assigns all channels according to the standard channel mapping shown in Table 7-1. If the input to Model 585 includes a program with an LFE channel assignment not in line with the assignments in Table 7-1, you need to assign the LFE channel manually using this menu. In multichannel audio, the LFE channel is bandwidth-limited, so to process the complete program audio properly, the LFE channel must be assigned correctly.

2.6.3   System Settings

In the `System Settings` menu, the options are `Display Mode` and `Brightness`.

Brightness

You can use the `Brightness` menu to adjust the intensity of the display. Changes you make in this menu are activated immediately, without pressing **Enter**.

Display Mode

Use `Display Mode` to choose the unit of adjustment displayed for pitch shifting: `frames`, `¢ (cents)`, or `% (percent)`.

Frame mode uses percent as the unit of adjustment. Adjustments are in increments of 0.1 percent. An additional setting of `+4.17%` is included, for pitch shifting presentations recorded at 25 fps but played back at 24 fps. The frame mode display also indicates when the setting is appropriate for pitch shifting presentations recorded at 24 fps but played back at 25 fps.

Cents represent musical cents, where 100 cents equals one semitone, and 1,200 cents equals one octave. Model 585 offers a shifting range of –282 to +242 cents.

Percent mode operates in the same manner as frame mode in all but two ways: it lacks the additional +4.17% setting, and does not indicate the values appropriate for conversions between frame rates.

Chapter 3
Time Scaling

Time scaling alters the duration of an audio signal, but leaves the perceived pitch unchanged from the original signal. Model 585 offers stand-alone, high-quality multichannel time scaling of short program segments by using the `Transport` menu.

At the standard 48 kHz sampling rate, Model 585 can store approximately 23 minutes of mono, 11.5 minutes of stereo, 6 minutes of four-channel, 4 minutes of 5.1-channel, or almost 3 minutes of eight-channel audio material. If the sampling rate changes, the storage capacity changes inversely: for example, a ten-percent reduction in sampling rate gives a ten-percent increase in storage capacity.

Model 585 also can be used in combination with other products to deliver continuous, online time scaling of longer programs.

3.1   Transport Menu

You can use the `Transport` menu to record an audio clip to internal memory and play back the clip time scaled at a rate within a range of ±15 percent.

In the `Transport` menu, the front-panel controls perform different tasks than in all other menus. This gives you access to the separate set of controls time scaling requires.

The `Transport` menu includes `Play, Record,` and `Program Select` menus. Entering either the `Play` or `Record` menu disables pitch shifting and reconfigures the four center buttons to trigger the function illustrated on the display above the button.

3.1.1   Play

Enter the `Play` menu to manipulate the time scaling of the clip stored in memory, or to simply play the clip, or any part of it.

When you enter the `Play` menu, you see:

```
  ■         00:00.00        🕐
  ■ /▶      %      In     Out
```

Status

Setup   ⌃   ⌄   Enter
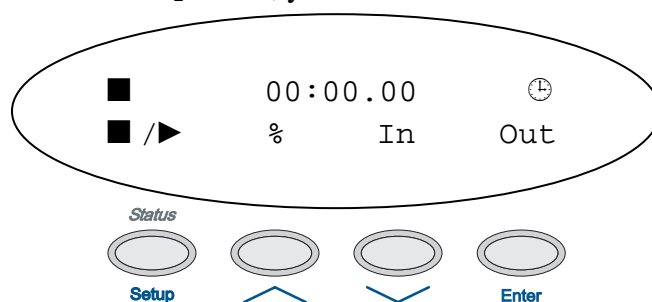
The upper line shows the transport status, the point in the stored clip where you last viewed the menu, and a clock symbol. When you enter the play menu, the transport status is in stop mode, as shown here. If you have not yet played any of the clip, the display reads `00:00.00,` as shown here. The clock symbol is there simply to remind you that Model 585 is in time scaling mode.

Note: The time displayed represents the scaled time value according to the current scaling percentage. If you change the scaling percentage, the time value changes accordingly.

The four center front-panel buttons are mapped to the functions displayed on the lower line of the display, as shown in Table 3-1. More details on button functions follow the table under the heading of each mapped function. Knob operation is also detailed.

Table 3-1 Play Menu Button Mapping

| Button	Play Menu Function	Result
Setup	Stop/Play	Stops/Starts playback of clip in internal memory.
⌃	Scaling Percentage	Enters the `%` menu, where you can adjust the scaling of the clip.
⌄	In Point	Moves the clip to the saved In point. To set the In point, press **Shift** + ⌄.
Enter	Out Point	Moves the clip to the saved Out point. To set the Out point, press **Shift** + **Enter.**

When a clip is stored in the Model 585 memory, the `Play` menu displays the scaled time value of that clip. If you change the scaling percentage, the length of the scaled clip changes to reflect that, and the `Play` menu display, In point, and Out point all update accordingly.

Stop/Play

In the `Play` menu, **Setup** acts as a Stop/Play toggle. When you enter the `Play` menu, Model 585 is in stop mode. When you press **Setup** in stop mode, playback of the clip starts. If no In point or Out point has been set, playback starts at the current transport position and continues until you press **Setup** again, or until the clip ends. When the clip ends, stop mode reactivates. When an In point or Out point is set, the Stop/Play command functions as described under those headings in this section.

Scaling Percentage

When you enter the `%` menu, you see:

```
Orig.           00:30.00
+10.0%          00:33.00
```

Status

Setup   ⌃   ⌄   Enter
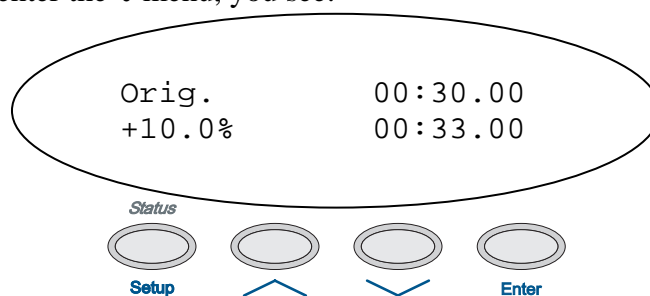
The screen values shown here are only an example.

The upper line shows the length of the original clip. The second line shows the scaling percentage and the resulting length of the scaled clip. You can adjust the scaling percentage in this menu using the arrow buttons, or by rotating the knob. As you change the scaling percentage, the display for the length of the scaled clip updates.

Scaling changes are applied immediately in this menu. If you are playing a clip, it continues playing at the new scaling percentage.

Note: You can only exit the Scaling Percentage menu by pressing **Esc.** If you select *Bypass*, playback stops.

In Point

The In point is the point in the audio clip when playback starts. If the transport position is at or before the In point when you press Play, playback begins at the In point. When the transport is at the In point, the display shows a dot before the word In: ·In.

To set or reset the In point to the current transport position, press **Shift** + ⌄. You can set the In point in either play or stop mode.

Note: If you need to reset the In point to a point earlier than the current In point, rotate the knob to move to a transport position before the In point and press **Shift** + ⌄ to reset the In point.


Out Point

The Out point is the point in the audio clip when playback ends. If the transport position is beyond the Out point when you press play, playback starts at that transport position and continues until the end of the clip, or until you stop playback. When the transport is at the Out point, the display shows a dot before the word Out: ·Out. When the transport is at the Out point, pressing play does not start playback.

To set or reset the Out point in the current clip, press **Shift** + **Enter.**

Knob Function in Play Menu

In the Play menu, you can use the knob to adjust the transport position in the recorded clip. As you move through the clip, the time display updates to show your position in the recorded clip. There are two ways to use the knob to adjust the transport position: rotating the knob by itself, and rotating the knob while holding down **Shift.**

When you rotate the knob by itself, the display shows ➤ or ◀ and a number indicating how fast you are moving through the clip. This rate reflects how fast you rotate the knob. This mode makes it easy to move short distances or locate a specific point in the clip. When you stop rotating the knob, the transport returns to stop mode.

When you hold down **Shift** while rotating the knob, you move through the clip according to the position of the knob compared to its position when you begin holding down **Shift.** In this mode, the display shows « or » and a number indicating how fast you are moving through the clip. Moving the knob 120° in either direction from the initial position brings you to the beginning or end of the clip. While in this mode, the audio level is lowered by 6 dB. The audio level returns to normal after the transport has stopped. Once you move the knob, as long as you hold the **Shift** button you continue to move at the speed and direction set by the position of the knob. When

you release **Shift,** the transport returns to stop mode at the position in the clip when you release **Shift.**

Holding **Shift** makes it particularly easy to move through the clip rapidly, but it also allows you to move slowly at a steady pace, without having to manually turn the knob.

3.1.2   Record

When you enter the `Record` menu, you see:

```
    ■              00:00.00
    ■/●                  P1
```

Status

Setup   ∧   ∨   Enter
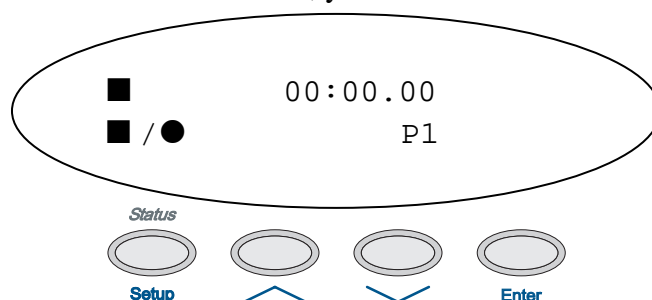
In this menu, the **Setup** button is a Stop/Record toggle. When you start a recording, the previous recording is erased, and the previous In point and Out point settings are cleared.

During recording, the top line displays ● and the elapsed recording time. When you stop a recording, the top line displays ■ and the total time of the clip just recorded. If you reach the maximum recording time, recording stops automatically.

Pressing **Esc** stops the recording and exits the `Record` menu. Selecting *Bypass* while in record mode stops the recording.

3.1.3   Program Selection

If the program configuration includes more than one program, use the `Program Select` menu to select the appropriate program to record. For a complete list of program configurations and the associated program numbers and channel assignments, see Table 7-1.

The second line in the `Record` menu displays which program records. In the example shown in Section 3.1.2, Record, `P1` indicates that Program 1 will record.

If the current program configuration includes more than one program, you can record any one of the programs or all the programs. For example, the `5.1 + 2` configuration contains two programs; you can record `P1` (program 1), `P2` (program 2), or `All` (both programs). In program configurations with more than two programs, you can select any single program, or `All`. Selecting `All` in any configuration means

all eight channels will be recorded, and the available recording memory is slightly less than three minutes at 48 kHz.

To select which program to record, use the arrow buttons, then press **Enter.**

3.2   Time Scaling of Longer Programs

While stand-alone time scaling using the `Transport` menu works well on short audio segments, it is performed offline, using the precision of the `Transport` menu controls. You can also perform online time scaling of a program of any length by using pitch shifting in combination with other storage and playback products. Online time scaling is discussed in Section 4.2, Using Pitch Shifting to Perform Online Time Scaling.

Chapter 4
Pitch Shifting

Pitch shifting, sometimes referred to in other documents as pitch scaling, alters the perceived frequency of the audio signal from that of the original signal without altering the duration of the signal.

Pitch shifting is most often used to compensate for a required change in playback rate of a program, so that the audio is reproduced at the original pitch in spite of the rate change. Use this equation to calculate the exact amount of pitch shift:

$$\left(\frac{\textit{input frame rate - output frame rate}}{\textit{output frame rate}} \right) \times \textit{100\%}$$

Frame mode makes it easy to convert material between different frame rates. The display indicates when the chosen pitch shift percentage is appropriate for conversions from 24 fps to 25 fps and from 25 fps to 24 fps.

4.1   Adjusting the Pitch Shifting Value
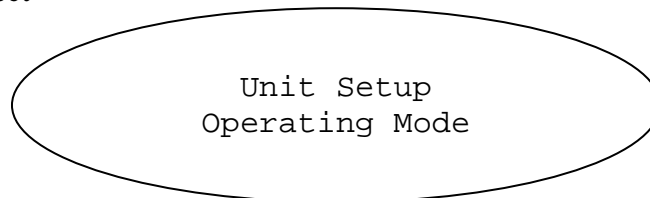
The pitch shifting function is active in every menu except `Transport`, unless Model 585 is in bypass mode. The pitch shifting value can be adjusted in two ways: in the `Pitch` menu, or in the main status menu.

Using the Pitch Menu

To adjust the pitch shifting value using the `Pitch` menu:

1. Select

```
            Unit Setup
          Operating Mode
```

and press **Enter.**

2. Use the knob or arrow buttons to scroll through the second line display until you get to `Pitch`, then press **Enter.**

3. Use the knob or arrow buttons to adjust the value on the second line.

    A flashing block displays as you change the value. The new value activates only when you press **Enter.** To leave the active value in effect and return the display to that value, press **Esc.**

Using the Main Status Menu

In the main status menu, you can adjust the pitch shift value by turning the knob. Pressing either arrow button scrolls to other status menu displays.

Caution: *Adjusting the pitch shift value in the main status menu activates that value change immediately, without pressing* **Enter**. *You do not have the option to press* **Esc** *to return to the original setting, so note what that setting is before rotating the knob.*

4.2   Using Pitch Shifting to Perform Online Time Scaling

You can use the pitch shifting function of Model 585 in combination with other equipment to deliver online time scaling of programs of any length.

To perform online time scaling:

1. Adjust the playback of the source program to match the timing requirements for final program delivery.

2. Use Model 585 to adjust the pitch shifting percentage.

    The amount of shift required is roughly opposite in value to the playback speed adjustment. For example, a program with a playback speed adjustment of +3.2% gets a pitch shift adjustment of –3.1%. Use the equation at the beginning of this chapter to calculate an exact value.

3. To maintain A/V sync, either add offsets in the timecode between the audio and video playback equipment, or add a delay to the video equal to the `Processing Delay` setting on Model 585.

    `Processing Delay` is an option under `Operating Mode.`

Chapter 5
Applications

Model 585 can be used to compensate for pitch changes in many applications, such as:

- When changing from 24 to 25 fps (for example, when converting a film source to video for broadcast or DVD release)
- When changing from 25 to 24 fps (for example, when converting a video source to 35 mm film for theatrical exhibition)
- When changing the duration of content to fit an allotted broadcast time slot

Model 585 can also be used in many equipment configurations. In this chapter we discuss specific examples for:

- Stand-alone audio time scaling
- Processing with synchronized video

5.1   Stand-Alone Processing

Model 585 can function as a stand-alone tool. Internal memory allows you to record and play back a short program. During playback, you can time scale stored audio. This setup is useful for processing short pieces of material that do not require synchronization to a visual program.

Figure 5-1 shows the stand-alone time scaling process.
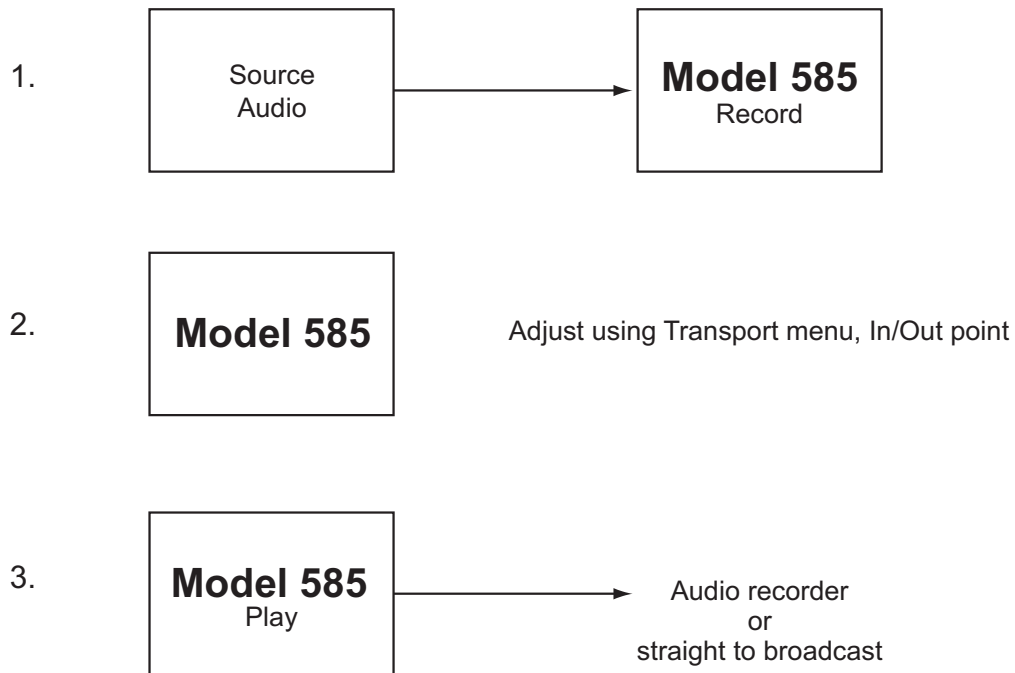
Figure 5-1 Stand-Alone Time Scaling

For details on how to operate Model 585 to perform stand-alone time scaling, see Section 3.1, Transport Menu.

5.2   Processing with Synchronized Video

To synchronize pitch shifted audio with its associated video program, the key element is compensating for the delay created by the audio processing. There are many ways this can be accomplished (for example, by using a single-pass approach, as shown in Figure 5-2, or by performing separate video and audio passes).
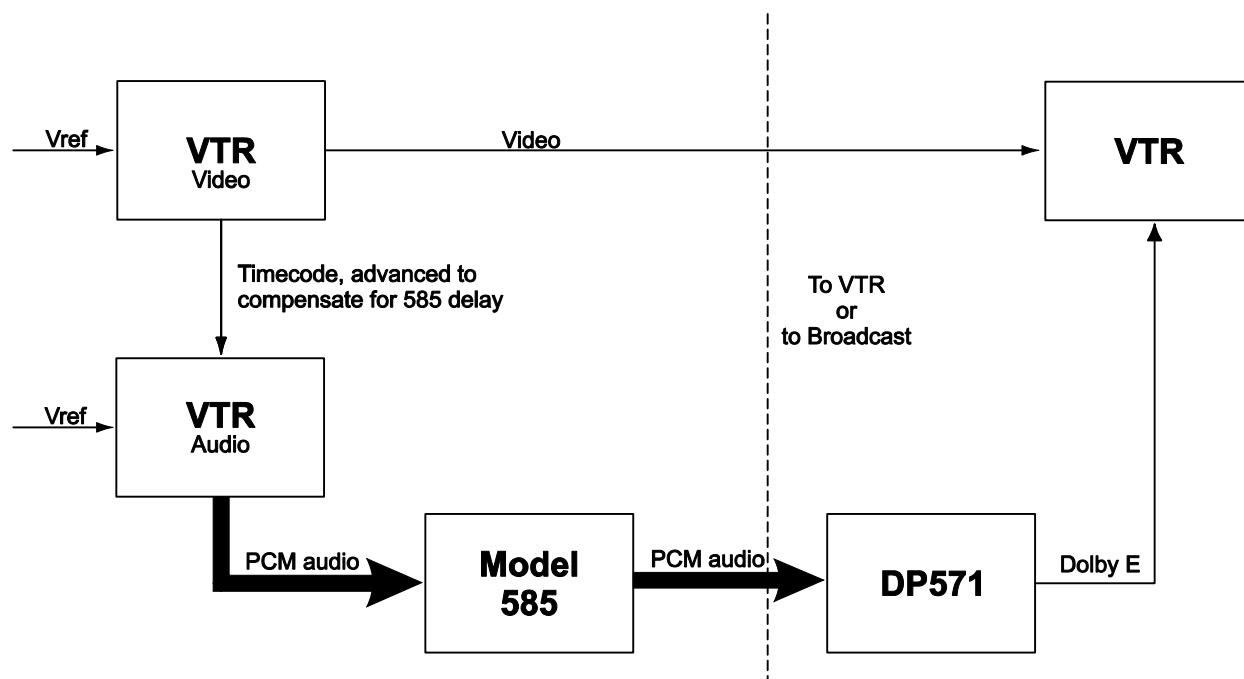
Figure 5-2 Single-Pass Pitch Shifting

In Figure 5-2, the source program video and audio are delivered separately. The playback VTR is used to adjust the speed of the program to conform to the requirements for final delivery. The playback VTR also sends timecode to the multitrack DTRS, which adjusts the speed of the multichannel audio. Model 585 compensates for the speed change by adjusting the pitch shifting to the opposite value, as discussed in section 4.2. By advancing the audio timecode by the same value as the delay setting on Model 585, you send synchronized audio and video in a program that is time scaled. The time scaled program is ready for either immediate broadcast, or recording onto a new master tape.

The amount of pitch shifting required is roughly opposite in value to the playback speed adjustment. See Section 4.1, Adjusting the Pitch Shifting Value, for details on how to set the pitch shifting value.

Chapter 6
Principles of Operation

Time scaling of audio has been accomplished previously using a variety of approaches. Model 585 differs from previous products in two fundamental ways:

- It provides phase-synchronous multichannel processing in addition to processing of simpler programs.
- It steps away from the weaknesses of the various previous approaches.

6.1   Multichannel Delivery

Multichannel audio is now standard in cinemas and is becoming more popular in broadcast programs as well. For the film, postproduction, and broadcast industries, time scaling is often necessary, but until now it has required a significant sacrifice in audio quality.

Model 585 delivers phase-synchronous multichannel time scaling of the highest quality. Phase-synchronous multichannel processing is essential in today's postproduction and broadcast industries because of the variety of replay environments found in listener's home. For example, a program broadcast or DVD made in Dolby® Digital carries the full 5.1-channel mix along with information that enables listeners to hear the best 5.1-channel, stereo, or mono mix possible from their audio setup.

If that same 5.1-channel mix is processed by three two-channel processors set to the same pitch shifting value, the loss of a common phase relationship between the audio channels can cause audible problems. The consumer with a complete home theater system hears six channels that are out-of-phase, while the consumer with a stereo or mono system hears a severely compromised audio program as the out-of-phase components are summed together.

6.2   Development of Time Scaling

Time scaling of audio was first achieved in the 1950s with the invention by Fairbanks, Everitt, and Jaeger[1] of a modified tape recorder. Their modified machine discarded and repeated sections of audio signal by using a rotating playback head, and became known as the *sampling* or *splicing* technique.

6.2.1   Time-Domain Solutions

Based on a time-domain representation of the signal, time-domain techniques are efficient and have been attractive for use in real-time implementations. The Fairbanks splicing technique was a very low-fidelity solution, producing a choppy product that sometimes obscured the content.

Lee[2] translated the splicing technique into a digital system in 1972, replacing the magnetic tape with a circular memory buffer and using read/write memory pointers in place of the rotating head. Further advances were made on this solution throughout the '70s, '80s, and '90s. These advances generally applied signal analysis to the splicing technique, making use of increased computer processing power during that time period. The solutions developed tended to focus on either speech or musical instruments, but not combinations of both.

The more recent solutions are significant improvements on the original splicing technique, but all time-domain based processors exhibit a characteristic drawback that at minimum makes the listener aware that the audio has been altered. Characteristic artifacts of these methods often include audible echoes, stutters and an overall loss of definition in the audio.

6.2.2   Frequency-Domain Solutions

In 1967, an alternative to time-domain techniques was introduced by Schroeder[3]. This method used either a Fourier transform or a filter bank to produce a frequency representation of the signal. A direct manipulation of the frequency components of the signal was followed by a transformation back into the time domain to achieve time scaling.

The technique introduced by Schroeder was developed for speech. Later modifications applied this approach to music, and variations were produced that could handle specific signal types, either music- or speech-oriented, but still not in combination.

---

[1] Fairbanks, G., W. Everitt, and R. P. Jaeger. "Method for Time or Frequency Compression-Expansion of Speech." *Transactions of the Institute of Radio Engineers, Professional Group on Audio* AU-2 (1954): 7–12.
[2] Lee, Francis F. "Time Compression and Expansion of Speech by the Sampling Method." *Journal of the Audio Engineering Society* 20, no. 9 (1972): 738–742.
[3] Schroeder, M., J. Flanagan and E. Lundry. "Band-Width Compression of Speech by Analytic-Signal Rooting." *Proceedings of the IEEE* 55 (1967): 396–401.

Frequency-domain solutions proved useful in certain situations, particularly where processing time was not a major factor. However, they also produced characteristic artifacts, and could only be used for a limited selection of material. Artifacts of frequency-domain solutions often include audible phase artifacts and the loss of definition of transient material.

6.2.3   Modeling-Based Solutions

In the 1980s, another approach was offered, this one based on signal modeling. Like frequency-domain solutions, signal modeling could be best used on less-complex material; different modeling applications tend to be effective on one or two types of material only. Processing time is also significant in signal modeling, effectively eliminating that as a real-time solution. Artifacts are still present, particularly if the audio is not strictly in line with the profile for which a specific modeling application was designed.

6.2.4   Choosing a Solution

Some desktop audio editing software packages include many variations of the time-scaling solutions just described. The labels given to the choices may provide clues as to which version may best apply to the audio you need to process, but every selection requires a certain amount of testing before deciding what works best.

Because of the variety and complexity in some program material, this time-consuming selection process has been, before Model 585, an exercise in choosing the least problematic option, rather than finding an optimal solution.

Model 585 eliminates the need for such a selection. It capably handles all combinations of speech, music, and special effects.

6.3   Intelligent Analysis

To perform time and pitch scaling, Model 585 performs a correlated multichannel version of the time-synchronous overlap-add technique, discarding segments of the input signal to achieve time compression, or copying and repeating segments of the input signal to achieve time expansion. This technique is chosen as the basis of the processing algorithm, as it is well suited to real-time application. The major improvement Model 585 provides over existing processors is in identifying which segments to discard or copy. To do this, Model 585 analyzes the psychoacoustic characteristics of a multichannel audio signal in both the frequency and time domains.

To make time scaling sound as natural as possible, Model 585 identifies each individual "auditory event" in an audio signal—such as a musical note, a drum hit, or the phonetic components of speech. Each event is isolated and time scaled individually. To accomplish this, Model 585 isolates audio components that have the

same characteristics (such as common onset, common harmonic properties, or in multichannel audio, common spatial location), and breaks down the complete sound into its component parts before discarding or adding samples.

Model 585 also takes advantage of psychoacoustic phenomena that have been studied previously for use in audio coding schemes. For example, at smaller time scales, the human brain focuses on a single auditory event at a given time. Therefore, processing a relatively large section of audio allows multiple auditory events to be analyzed and identified.

As the complexity of the audio increases, Model 585 identifies the most prominent element at any given moment and adjusts the time scaling or pitch shifting accordingly. It finds both the most prominent auditory events, as well as lesser components that occur simultaneously. (The `Sensitivity` setting controls the degree to which lesser components are analyzed, as discussed in Section 2.6.2). Also, by identifying the boundaries of events, which may exist in both the frequency and time domains, crossfades between spliced sections can be performed with greatly improved quality when compared to the methods that are used elsewhere.

Chapter 7
Reference Data

7.1 Channel Mapping

Table 7-1 Channel Mapping according to Program Configuration
The program number (1–8) identifies the individual programs associated with each channel.
So, the entry 2C/2S *means "Program 2, Center channel/Program 2, Surround channel."*

Program Configuration	Channel			
	1/2	**3/4**	**5/6**	**7/8**
5.1 + 2	1L/1R	1C/1LFE	1Ls/1Rs	2L/2R
5.1 + 2 × 1	1L/1R	1C/1LFE	1Ls/1Rs	2C/3C
4 + 4	1L/1R	1C/1S	2C/2S	2L/2R
4 + 2 × 2	1L/1R	1C/1S	3L/3R	2L/2R
4 + 2 + 2 × 1	1L/1R	1C/1S	3C/4C	2L/2R
4 + 4 × 1	1L/1R	1C/1S	4C/5C	2C/3C
4 × 2	1L/1R	3L/3R	4L/4R	2L/2R
3 × 2 + 2 × 1	1L/1R	3L/3R	4C/5C	2L/2R
2 × 2 + 4 × 1	1L/1R	3C/4C	5C/6C	2L/2R
2 + 6 × 1	1L/1R	4C/5C	6C/7C	2C/3C
8 × 1	1C/2C	3C/4C	5C/6C	7C/8C
5.1	1L/1R	1C/1LFE	1Ls/1Rs	None
4 + 2	1L/1R	1C/1S	None	2L/2R
4 + 2 × 1	1L/1R	1C/1S	None	2C/3C
3 × 2	1L/1R	3L/3R	None	2L/2R
2 × 2 + 2 × 1	1L/1R	3C/4C	None	2L/2R
2 + 4 × 1	1L/1R	4C/5C	None	2C/3C
6 × 1	1C/2C	3C/4C	5C/6C	None
4	1L/1R	1C/1S	None	None
2 + 2	1L/1R	None	None	2L/2R
2 + 2 × 1	1L/1R	None	None	2C/3C
4 × 1	1C/2C	3C/4C	None	None
7.1	1L/1R	1C/1LFE	1Ls/1Rs	1BLs/1BRs
7.1 Scrn	1L/1R	1C/1LFE	1Ls/1Rs	1BLs/1BRs

> **Note:** In multichannel audio, the LFE channel is bandwidth-limited, so to properly process the complete program audio, the LFE channel must be assigned according to standard channel mapping.

7.2   Specifications

Time Scaling/Pitch Correction Process
Dolby proprietary, real-time processing

Channel Modes/Program Configurations
Processes up to eight discrete channels of audio; supports all Dolby® E program configurations, including 5.1, 5.1 + 2, 4 × 2, 8 × 1

Pitch Shifting Range
±15% (0.1% increments), or –282 cents to +242 cents (in 2-cent increments)

Time Scaling Range
±15% (0.1% increments)

Time Scaling Storage Limits
Internal memory:
    mono: 23 min 11 sec
    stereo: 11 min 35 sec
    4 channels: 5 min 48 sec
    5.1 channels: 3 min 50 sec
    8 channels: 2 min 54 sec

Audio Sampling Rates
48 kHz, ±15% (40.8 to 55.2 kHz)

Frequency Response
20 Hz to 20 kHz, ±0.25 dB

Distortion
<0.01% at 1 kHz
<0.02%, 20 Hz to 20 kHz

Dynamic Range
>110 dB

Delay
400 to 480 ms, user-selectable

Digital Audio Inputs
Four BNC female connectors with loop-through, 75Ω, unbalanced, signal levels per AES-3ID-1995/SMPTE 276M specifications

Digital Audio Outputs
Four BNC female connectors, 75Ω, unbalanced, signal levels per AES-3ID-1995/SMPTE 276M specifications

AES Reference Input
BNC female with loop-through, 75Ω, unbalanced, signal levels per AES3-ID

Audio Word Length
Up to 24 bits

Serial Remote Control Input
Rear: RS-232, 9-pin female D-connector

Auxiliary Ports
Rear: Two 9-pin female D-connectors, RS-422, full duplex

Metadata Input/Output Ports
Rear: 9-pin female D-connector, 115 kbps, pinout per SMPTE 207M (RS-485)

General Purpose Input/Output (GPI/O) Port
Rear: 9-pin female D-connector, 0 to 5 V TTL levels

Ethernet Port
Rear: RJ-45 female connector, 10Base-T

Power Requirements
90 to 264 VAC, 50 to 60 Hz, 15 W maximum; designed to operate from a centrally switched power source

Dimensions and Weight

2-U rack-mount: $88 \times 483 \times 324$ mm ($3.5 \times 19 \times 12.75$ inches)
Net: 6.5 kg (14.25 lb)

Environmental Conditions

Operating: 0° to 50° C (32° to 122° F), natural convection cooling
0 to 98% relative humidity (non-condensing)
Non-operating: -20° to $+70^{\circ}$ C (-4° to $+158^{\circ}$ F)